PSP NITRO


CFG OPS LFO ENV ADS MOD LIB GLB
IN
PREDEF
dB
OP1
OP2
OP3
OP4
OUT
+0.0dB
100.0%
-10.8dB
ON
on
solo
m/s
level
FBACK
DEPTH
FREQU
RESON
in
mix
out
power
dry
wet

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Forward

Thank you for your purchase of the PSP Nitro, one of the most powerful and flexible multimode filter plug-ins available! You can use this plug-in to shape your audio signal in nearly unlimited ways. Our flexible modulation system allows nearly every parameter to be modulated by either internal modulators or external controllers. Finally PSP Nitro offers a complete preset management system. All of this is included in a functional and attractive interface.

PSP Nitro is a very powerful and flexible plug-in. With power and flexibility a certain level of complexity is inevitable. Although we have tried to design it to be as user friendly and intuitive as possible, we recommend that you spend some time perusing the manual to become familiar with how it operates and to get a sense of PSP Nitro's potential. We have tried to include complete descriptions and tutorials of how to operate every section of our plug-in; please take advantage of it!

We hope that you find this plug-in useful and remember—creativity and experimentation are the name of the game!

Thank you,

Your PSPaudioware.com team

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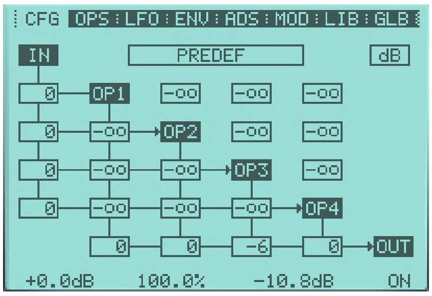
An Overview of PSP Nitro's Architecture

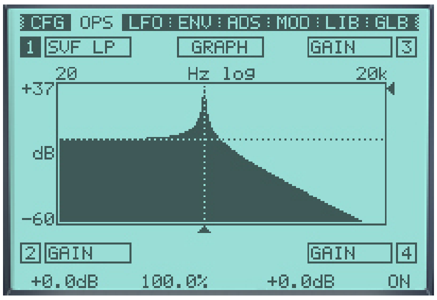
PSP Nitro offers too many features and parameters for the editor window to display everything in a single screen. To accommodate PSP Nitro's many options in a reasonably sized editor window, the fundamental modules that comprise Nitro's "sound engine" are permanently displayed on the "face" of the editor, while those parameters mostly used for configuration, file management, or the like, are accessed through the "LCD" screen in the center of the interface. Each module and screen will be explained in detail in the following sections, but to help you understand the signal flow within PSP Nitro, we'd like to offer a short description of the architecture and structure of the plug-in.

The most fundamental module in PSP Nitro is the "Operator" shown below. The Operator is assigned one of PSP Nitro's various filter types. When audio passes through the Operator, the selected filter "operates" on the audio—hence the name for the module. PSP Nitro offers four such Operators, meaning four completely different "operations" lay at your fingertips! Once you grasp the concept that PSP Nitro consists of four Operators that process your audio, everything else should fall into place easily. The parameters available on the GUI for the Operators are described in the **Main Editor Controls** section.

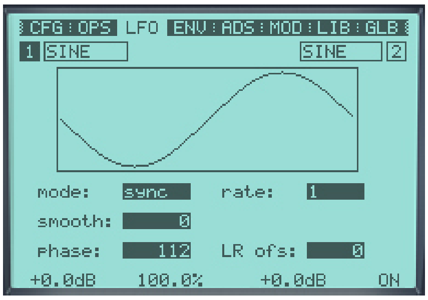
In order for your audio to reach these Operators, you first configure how the audio will travel through the various Operators in the configuration screen, which is represented in the CFG tab of the LCD screen. The CFG screen is shown below. Here you can set up any routing between the four Operators you wish. You can have the signal running through the various Operators sequentially, in parallel, and in any other sort of unique and creative order your heart desires. As you can see above, your audio signal starts at the IN box, and winds its way through your routing configuration until it reaches the OUT box. How to create routings is explained below in the **CFG** section.



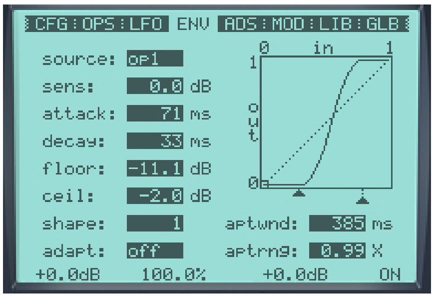
Once you have determined your audio path, your audio will pass through each Operator you have placed in the signal path in the order you configured. You assign filters to each operator in the OPS tab of the LCD screen, shown below. Here you can choose one of PSP Nitro's 18 filters/effects for each operator. The center of the OPS screen displays either the available parameters or a graph of the current settings of the selected Operator. How to use this screen is explained in detail in the **OPS** section.



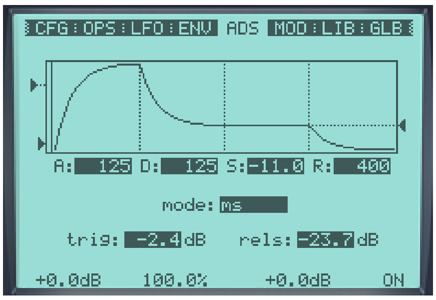
While each Operator in the signal path processes the audio signal, the signal can be modulated through the two low frequency oscillators. These LFOs are available in the LFO tab of the LCD screen. The LFO screen is shown below. The LFO screen shows a graph of the LFO curve, and the current parameter settings, for the selected LFO. How to set the LFO is explained in the **LFO** section.



Another modulation option in addition to the low frequency oscillators, the envelope detector can analyze the signal from the input, from the output of any of the Operators, or the output signal. The envelope detector is accessed from the ENV screen, shown below. The envelope detector can shape the selected output in various ways, as described in the **ENV** section.

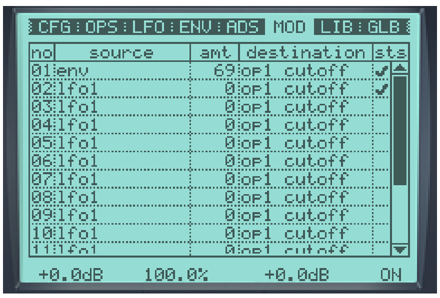


Finally, the third modulator offered is PSP Nitro's ADSR (Attack Decay Sustain Release) envelope. This envelope is located on the ADS tab, shown below. The ADSR envelope can be triggered by any modulation source such as the envelope detector output or a MIDI note on message. How to use the ADSR generator is explained in the **ADS** section.

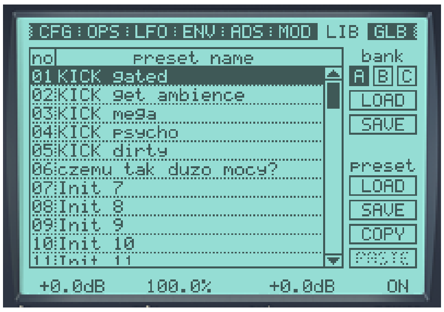


There are a few more sections of PSP Nitro to describe before we move on to the individual sections.

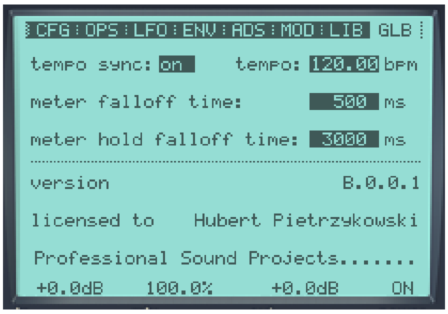
PSP Nitro contains a highly flexible modulation routing system that you can access from the MOD tab. The modulation screen is shown below. Most MIDI messages, as well as Nitro's envelopes, can be chosen as modulation sources. The modulation sources can modulate nearly all of Nitro's parameters and routing connections. How to use the modulation matrix is explained in the **MOD** section.



PSP Nitro uses its own preset management system, giving PSP Nitro consistent preset handling regardless of host application or plug-in format. The preset management screen is accessed from the LIB tab. The LIB screen is shown below. Using PSP Nitro's preset management functions is explained in the **LIB** section.



There are a number of parameters that do not affect the processing of the plug-in, but do affect Nitro's operation. These parameters, along with your license information, are contained in the Global screen, accessed from the GLB tab, shown below. An explanation of these parameters is given in the **GLB** section.



Lastly, the Mixer section contains the input and output gain, mix control, power (bypass) button, and level meters as shown below. These controls function like a hardware mixer into which PSP Nitro is "plugged in". This section allows you to fine tune the levels of PSP Nitro to the audio track. This section is described in more detail in the **Main Editor Controls** section.

Main Editor Controls

The graphic user interface of PSP Nitro gives you instant access to the Operator and Mixer sections of the plug-in. Regardless of which tab of the LCD display you are working with, you can always adjust these.

The controls on the main editor have been divided into two sub-sections, the Operator module, describing the controls in each Operator, and the Mixer section, explaining the controls in the mixer.

Operator Module

PSP Nitro offers an identical set of knobs, buttons, and sliders to control each of the four Operator modules. This allows the user to instantly manipulate the parameters of each module in realtime without needing to dig through tabs in the LCD display. The function of these knobs and buttons are described below.

on button

This button turns on the Operator. If this button is turned off, the Operator is bypassed.

solo button

The solo button sends the processing of only the selected Operator to the output, in effect "soloing" the selected Operator. This is useful when you want to only listen to one Operator while adjusting its parameters.

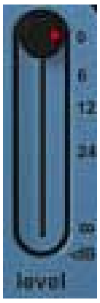
m/s button

This button changes stereo field of the Operator from the usual left/right stereo spread to middle/side. It allows for the center signal to be processed differently than the side signals. You can use this to experiment with creative LFO/filter interactions. For example, you could set an Operator to middle/side mode, modulate one parameter of a filter with the LFO, and one channel of the LFO would modulate the middle signal, while the other channel of the LFO would modulate the side signal.

parameter knob

As explained in the previous section, each Operator that you wish to use needs to be assigned one of the available filters or effects. You do this in the OPS tab of the LCD display, which is described later in the **OPS** section. Each filter/effect offers either one or more user-adjustable parameters, depending on the specific filter/effect you have assigned. The main parameters of the assigned filter are always accessible via these two Parameter Knobs of each Operator module. The name of the parameter assigned to each knob appears in the red display beneath the knob. You can adjust the parameter by clicking on the knob and moving your mouse. When adjusting a parameter, the red display will change to show you the current value of the parameter; when you are done adjusting the parameter, the knob will revert to the name of the assigned parameter.

level slider



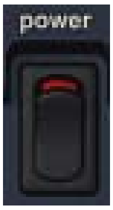
This slider allows you to adjust the output level of the Operator. You can use this to adjust the balance between the various Operators you have assigned.

Mixer Section

This section contains general controls that govern the level and mix of PSP Nitro with the source audio.

The current value of each control is displayed at the bottom of the LCD display directly above each control, as shown above. Below is a brief description of the elements of the Mixer Section.

power button



PSP Nitro's power button functions as a bypass switch. When switched off, this button will bypass the entire plug-in. When the power button is switched on, the plug-in will function normally.

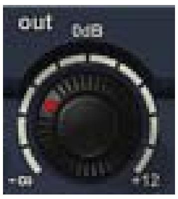
in knob

The *in* knob acts as an input trim, allowing you to cut or boost the signal level entering PSP Nitro. If you turn the in knob completely to the left, the input to be completely cut off (-∞dB). If you turn the in knob completely to the right, the input signal will be boosted by +12dB.

mix knob

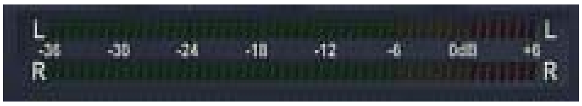
You can use the *mix* knob to adjust the dry to wet balance of the output. Turning the mix knob completely to the left (dry) will only output the original audio signal, while setting the mix knob completely to the right (wet) will output only the processed signal.

out knob



This knob acts as an output trim, allowing you to reduce or boost the signal level PSP Nitro outputs to the audio channel. The *out* knob allows for the output to be completely cut off (-∞dB), or boosted by up to +12dB.

peak level meters



The peak level meters show the momentary level and peak-hold level. The falloff time of the peak level meters can be adjusted on the GLB tab of the LCD display. This is discussed in the **GLB** section.

LCD Display Screens

The LCD display in the center of the interface contains tabs for the various modules and parameters that allow you to get into the “guts” of PSP Nitro. This is what gives the plug-in its power and flexibility. At the same time, offering this level of flexibility also results in more complexity. These following sections will guide you through each tab of the LCD display.

As shown below in **Figure 1**, the very top line and bottom line of the LCD display will always present the same information, regardless of which tab is showing. The very top line will always display a list of all of the available tabs. The name of the tab currently being displayed will appear highlighted in the same color as the background; the names of other tabs will all appear darkened. The bottom line will always display the values of the *in, mix, out*, and *power* buttons.

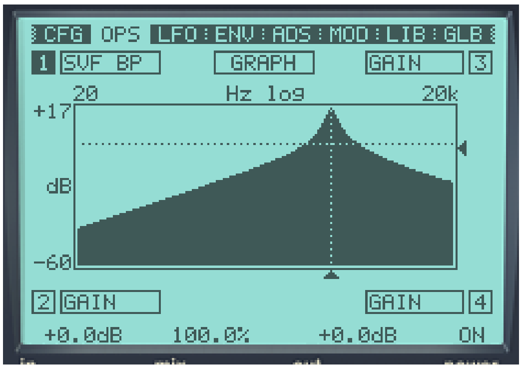


Fig 1: The top line of the LCD display always shows the names of all the tabs, with the current tab highlighted; the bottom line always shows the parameter values of the mixer section.

CFG Section

The CFG tab contains the routing matrix for PSP Nitro. You can connect the input, output, and four operators to each other using any imaginable combination of routings. The IN (input), OP1 (Operator 1), OP2 (Operator 2), OP3 (Operator 3), OP4 (Operator 4), and OUT (output) make up the diagonal axis of the matrix, and the boxes represent "switches" that guide the input signal along the signal path you wish. Lines between the boxes represent the signal path, with arrows representing signal direction. **Figure 2** shows an example CFG screen.

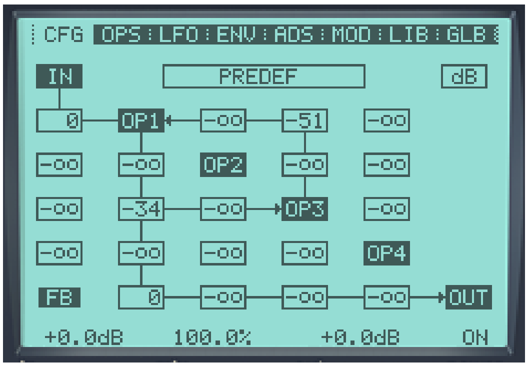


Fig. 2: PSP Nitro allows for nearly unlimited routing possibilities within its matrix. In the CFG screen above, the lines and arrow indicate the signal path, As you can see, the signal travels from the input, through OP1, then loops through OP3, and finally end at the output.

In addition to the routing matrix, the CFG screen contains the PREDEF button, which offers you access to a number of predefined routings, the dB/VAL button, which changes the values inside the routing boxes from dB to relative values between 0 and 100, and if your routing results in a feedback loop, the Feedback Indicator (FB) button will flash in the bottom left corner.

Basic Routing Matrix Operation

In dB mode, each box will show the gain of the connection from one operator (or input) to another (or output) and can be either a negative infinity symbol (-∞) or a number from –99 to 0. The location of the box in the routing matrix is determined by the source and destination of the connection: the source determines the column and the destination determines the row.

The negative infinity symbol indicates that the box does not route the signal at all. The number represents the gain reduction (attenuation), being applied to the signal at that box, from –99dB of attenuation to 0dB of attenuation. Generally, a value of 0dB when you are creating a basic routing offers the maximum signal level (no attenuation), but when you start creating complex routings you may want to attenuate the signal before your signal enters various points in the signal path. Don't worry, this is explained in more detail below.

To route the audio signal between any two sources in the matrix, you must change the value inside the box at the intersection of the horizontal and vertical planes of those two sources. This will create a connection between the two sources. You change the value inside the box by clicking inside that box and dragging the mouse until you reach the desired value. As soon as you change the value from off (-∞) to a number, a cable will appear connecting this box to the operators, input, or output that are on the same horizontal and vertical plane as that box.

If that sounds complicated to you, don't worry—it sounds more complicated than it is. Let's go through some examples of using the CFG matrix so you get the hang of it.

Creating a basic connection between IN, OP1, and OUT

We'll start by creating the most basic routing matrix—cabling the input through a single Operator to the output. **Figure 3** shows a matrix in which there are no connections at all.

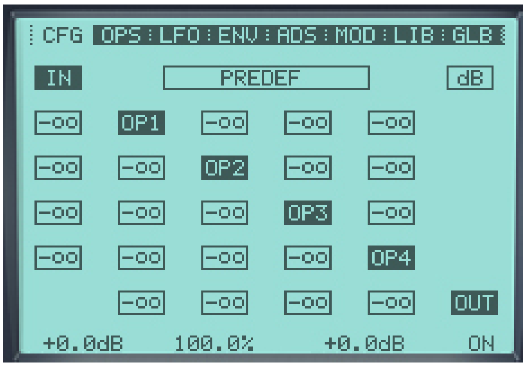


Fig. 3: A CFG matrix with nothing routed anywhere.

The first thing we want to do is to connect the input (IN) to Operator 1 (OP1). In order to do this, we need to find the box in which the horizontal and vertical planes between IN and OP1 intersect. That is the box directly below the IN box, and to the left of OP1. Click on this box and raise the value to 0dB. You will immediately notice a cable appear connecting box sources through the box you have adjusted, as shown in **Figure 4**.

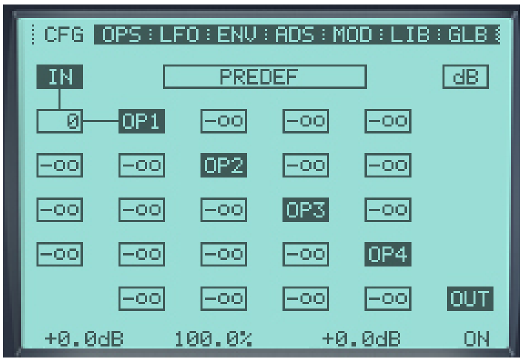


Fig. 4: By adjusting the value of the box above, the audio signal is routed between the input and Operator 1.

Now that we have the input signal going into Operator 1, we want to send it to the output. To do this, we need to find the box into which the horizontal and vertical planes between OP1 and OUT intersect. This would be the box in the bottom row underneath OP1, four boxes left of OUT. Click in that box and raise the value to 0dB. Notice that a line now appears between all the boxes between OP1 and OUT. Even though most of these boxes show -∞, they do pass the signal traveling along the cable we have just set up. It is only the box positioned in the corner (in the column of the source and in the row of the destination) that affects the connection gain. You will also notice when the cable connects to the output, it ends in an arrow to show you the direction of the signal path. **Figure 5** shows the complete connection.

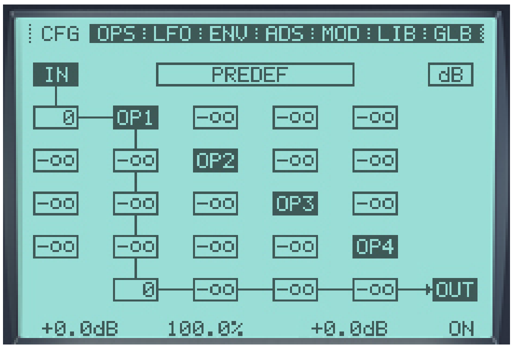


Fig. 5: By adjusting the box at the intersection of the horizontal and vertical planes between OP1 and OUT, you have connected the input of PSP Nitro, through Operator 1, to the plug-in's output.

That wasn't so complicated, was it? Next, we'll show you a more complex routing in which one operator loops back through another before the output.

Creating A Complex Routing with a Feedback Loop

Ok, now we're getting serious! We want to route our audio signal through all four Operators, and we want our signal, once it reaches OP3, to be routed back through OP2. You should already understand the basic concept of adjusting the box at the intersection of the horizontal and vertical planes between the sources you wish to connect. This example is really no more complicated than the first example; it simply requires more forethought to keep the signal level from going crazy.

Again, let's start with a completely empty routing matrix, just like we did for the last example. **Figure 6** shows this empty matrix.

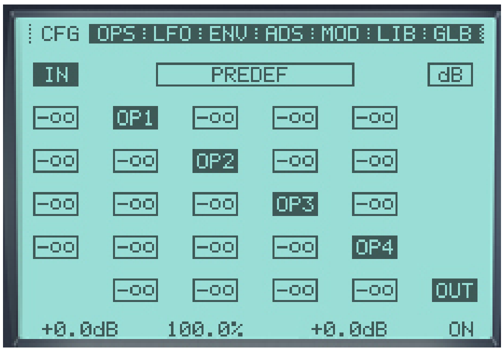


Fig 6: A matrix with no routings set, just like **Fig. 3**.

The first thing we want to do, just like the last example, is to connect the input (IN) to the first Operator (OP1). Click on the box and raise its value to 0dB, as shown in **Figure 7**. You will see a cable drawn between IN and OP1.

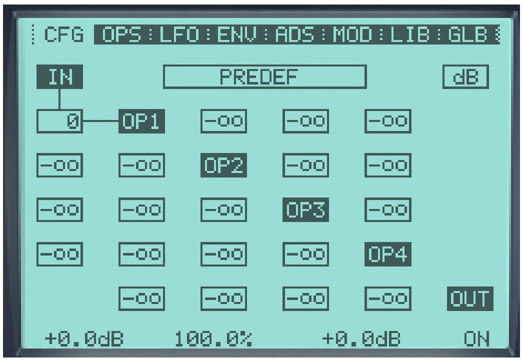


Fig. 7: IN and OP1 are now routed together, as in Fig. 4.

Here is where this routing starts diverging from our first routing exercise. We want to route our first Operator (OP1) to our second Operator (OP2). Using the same technique we used above, we will find the box that intersects the horizontal and vertical planes of both OP1 and OP2. In this case, there are actually two boxes that do this—third box on the top row of the matrix, or the second box on the second row of the matrix. We will use the second box on the second row of the matrix. We will discuss why we chose this box below—for now, adjust the value of this box to 0dB. As **Figure 8** shows, you will now see the cable indicating that OP1 and OP2 are cabled together, and the arrow indicating that the signal is flowing from OP1 to OP2.

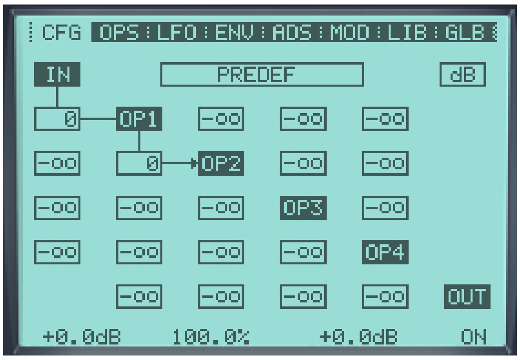


Fig. 8: By adjusting the value in the second box in the second row to 0dB, we have now routed OP1 and OP2 together. The direction of the arrow indicates that the audio signal will pass from OP1 to OP2.

Now that our audio signal is passing from the IN(put) to OP1, then OP2, we want our signal to continue on through OP3. Using the same technique we have been using, we will adjust to 0dB the lower of the two boxes at the intersection of the horizontal and vertical intersections of OP2 and OP3. **Figure 9** shows us that our audio signal is now routed from OP2 to OP3, with the arrow showing us the direction of our signal.

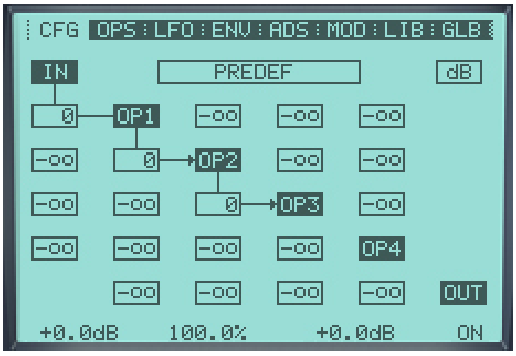


Fig. 9: By adjusting the value in the third box on the third row to 0dB, we have routed OP2 to OP3. The arrow indicates the direction of the signal flow, from OP2 to OP3.

Time to use the PSP Nitro routing matrix for some special effects! Lets presume that after processing our audio signal with OP3, we want to re-route our audio signal back through OP2 for even more processing. Easy! When we've been routing our signals forward in the matrix we have been using the boxes underneath our sources. In order to route our audio backwards we use the boxes above our sources. So to re-route our audio signal from OP3 to OP2, we will use the box that is at the intersection of the horizontal and vertical planes above OP3 and OP2. **Figure 10** shows OP3 and OP2 cabled together, with the direction of the arrow showing the audio signal direction. It is also possible to create the feedback routing from operator's output to its input by clicking and dragging on the operator rectangle itself.

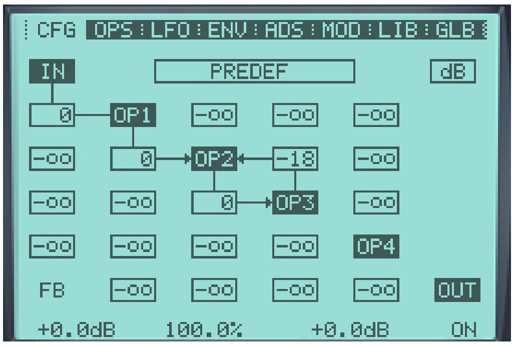


Fig. 10: This time, we selected the upper box that intersects the horizontal and vertical planes of OP3 and OP2, creating a loop. The direction of the arrows indicates that the audio signal is routed first from OP2 to OP3, then from OP3 back to OP2.

Notice in Figure 10, in the lower left of the screen, the flashing feedback loop (FB) button. This button warns you that by creating a loop between two Operators, you run the risk of the repeat processing overloading your signal. In order to reduce that risk, you can attenuate (reduce the level) of your audio signal as it returns to OP2. In our example, we are reducing the level of the audio by 18dB on the return loop from OP3 to OP2 in order to keep the level under control, as indicated by the value of –18 in the box.

Now we want our audio coming out of the OP2 to continue on to OP4. To continue our signal routing forward, we will choose the intersection of the horizontal and vertical planes between OP2 and OP4 that is underneath the sources, and adjust the value to 0dB. As **Figure 11** shows, our audio signal has now been routed past our feedback loop into OP4.

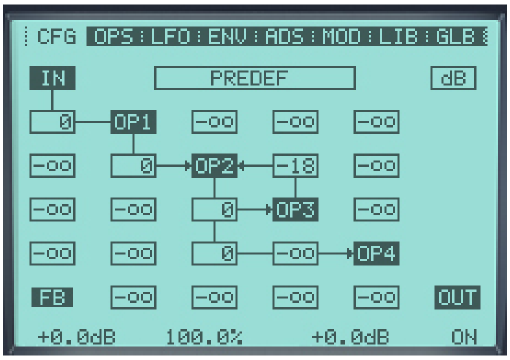


Fig. 11: Adjust the value to 0dB of the fourth box in the third row to route the audio signal from OP2 to OP4. The direction of the arrow shows you that the routings are now going forward again.

To finish our routing, we'll connect OP4 to the OUT(put), by adjusting the value in the box beneath OP4 and to the left of OUT to 0dB. **Figure 12** shows our complete routing.

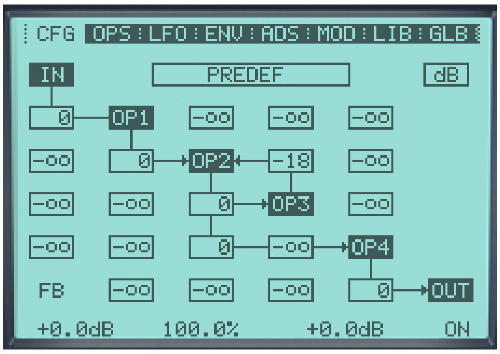


Fig. 12: By routing OP4 to our OUT(put), we have finished our routing exercise!

Operators In Parallel

We don't want you to get the idea from the two previous exercises that you can only use the Matrix to route the Operators together serially. This final exercise builds on the previous one, with the addition of a parallel signal chain to the fourth Operator (OP4).

Starting with the result of our previous exercise (as shown in Figure 12), lets add a new audio path direct from IN to OP4. By now the technique should be familiar—find the box at the horizontal and vertical intersection of IN and OP4 (the last box in the first row) and adjust its value. Since we want to be careful of signal overloads, and keep our serial signal loudest, lets attenuate the value to –25dB. We can see our resulting routing in **Figure 13**.

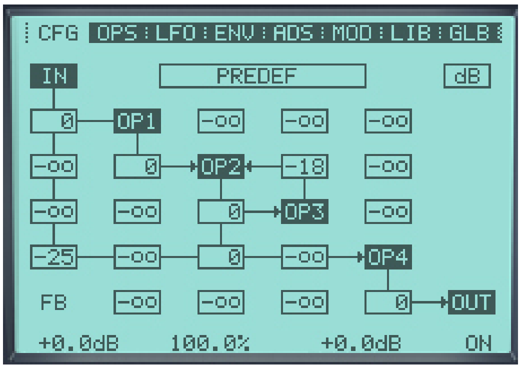


Fig. 13: The audio signal is now routed both through all for Operators in serial, and directly from IN to OP4 in parallel to the above routing.

That's all there is to it! Hopefully these exercises give you an idea as to the potential of the CFG matrix, and the routing possibilities of PSP Nitro. Your creativity is the only limit!

The PREDEF button

For those occasions that you do not want to set up your own routing matrix, the PREDEF button is there for you. When you click on the PREDEF button above the routing matrix, the screen will switch to a display of predefined signal routings. You can click on any of the routings to select it, the vertical bars to move forward or backwards through the 11 routings, or click the EXIT button to return to the routing matrix. **Figure 14** shows the PREDEF routing screen.

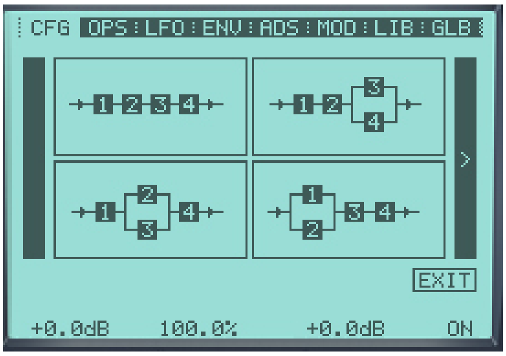


Fig. 14: The PREDEF routing screen. Click on any of the 11 routings to instantly set up your CFG matrix.

The 11 PREDEF routings do not offer you nearly the possibilities your have at your fingertips if you create your own routings in the CFG matrix, but using the PREDEF button to select a routing is fast and convenient.

OPS Section

The OPS section is the heart of PSP Nitro. This is where the four Operators that actually process your audio signal can be selected and adjusted. The four corners of the OPS screen each contain one of the four Operators. The number indicates which Operator number is in that corner, and the text box next to the number contains the processor type for that Operator. When you click the mouse on one of the numbers next to each Operator, you select that Operator for display and/or editing in the center of the display. **Figure 15** shows you the OPS screen.

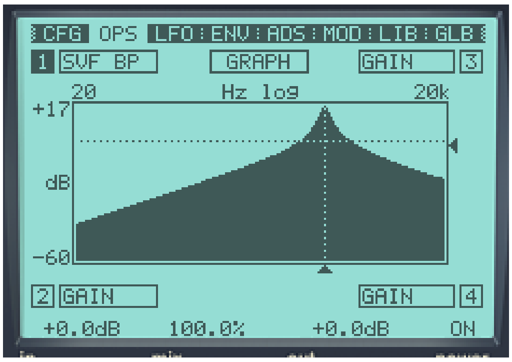


Fig. 15: The OPS Section of the LCD Display. Each Operator is listed in a different corner of the display. The highlighted number "1" above indicates that it is Operator 1 that is displayed in the center of the screen.

Clicking and moving the either or both of the arrows on the side of the graph has the same effect as clicking and moving the two parameter knobs in the Operator Module of the main editor. In fact, if you have the OPS Section showing on your LCD display and you adjust the parameter knobs of the selected Operator, the graph will display your parameter knob adjustments in real-time.

Notice the text box with the word 'GRAPH' centered above the graph. If you click in this box, the graph will change to a text display of the editable values of the Operator, as shown in **Figure 16**. Clicking in a parameter's text box and dragging the mouse up or down will raise or lower the value of the parameter. The VALUE view is useful for when you want to make precise measurements.

Some of the simpler processors, such as Gain, the Panner, and so on, do not offer a graph view, and the only view available will be a value display.

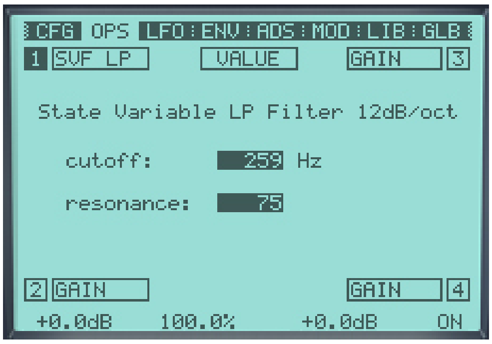


Fig 16: If the Operator being displayed offers a graph mode, clicking on the text box labeled GRAPH will change the view to VALUE, in which the editable parameters and their values will be displayed as text.

Types of Operators
When you click on the text box displaying the Operator type, the display changes to a text list of all 17 available Operator types. **Figure 17** shows you this text list. You can choose any processor type for any of the four Operators. You can choose the same type of processor for all four Operators, different processors for each operator, and any combination in between. There are no restrictions.

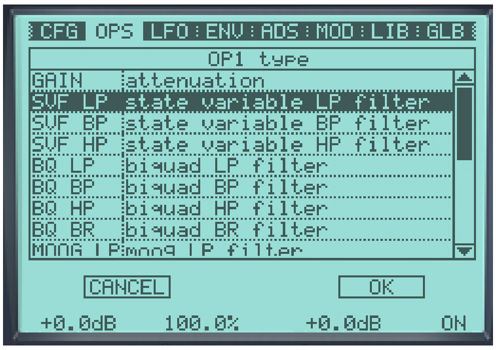


Fig. 17: When you click on the Operator name in the OPS display, you are presented with this listing of all 17 available Operator types.

Below is a brief description of the 17 Operator types, and their editable parameters.

State Variable Filters
PSP Nitro offers three separate 2-pole, double-sampled, 12dB/octave state variable filters modeled after the analog state variable filters used in classic synthesizers. Double sampling was used to make the filter high frequency response more like that of analog filters. Each filter offers parameters for the *cutoff* frequency of the filter and *resonance* around the cutoff frequency. The cutoff is adjustable between 20Hz to 20kHz, while the resonance can vary from no peak around the cutoff frequency to self-oscillation. The filters have slight internal non-linearity, which results in creation of harmonics, much like analog filters.

You have three filter options:

- High-pass filter **[SVF HP]**—only signals above the cutoff frequency pass through the filter.
- Low-pass filter **[SVF LP]**—only signals below the cutoff frequency pass through the filter.

- Band-pass filter **[SVF BP]**—only signals within a certain range around the center frequency pass through the filter.

Bi-Quad Filters

Another set of digital 2-pole filters derived from analog prototypes. With bi-quad filters, the maximum resonance value results in a high peak around the cutoff frequency, but unlike the state variable filters, no self-oscillation occurs. Each filter offers parameters for the *cutoff* frequency and *resonance*. You have four filter options:

- High-pass filter **[BQ HP]**—only signals above the cutoff frequency pass through the filter.
- Low-pass filter **[BQ LP]**—only signals below the cutoff frequency pass through the filter.
- Band-pass filter **[BQ BP]**—only signals within a certain range around the center frequency pass through the filter.
- Band-reject (also known as a Notch) filter **[BQ BR]**—the signal around the center frequency is rejected, and the rest of the signal allowed to pass through.

Moog Filters

The legendary Moog synthesizers were famous for the unique sound and structure of their 4-pole, 24dB/octave ladder filters. Just as with the original Moog filters, PSP Nitro's moog filter, we implemented this filter as a cascade of four 1-pole filters with a feedback path from the last filter back to the first one. Each filter offers parameters for the *cutoff* frequency and *resonance*. You have three filter options:

- High-pass filter **[MOOG HP]**—only signals above the cutoff frequency pass through the filter. Unlike the low-pass filter, this filter has a slope of 6dB/octave.
- Low-pass filter **[MOOG LP]**—only signals below the cutoff frequency pass through the filter. The filter slope is 24dB/octave.
- Band-pass filter **[MOOG BP]**—only signals within a certain range of the cutoff frequency pass through the filter. Unlike the low-pass filter, this filter has a slope of 6dB/octave.

Comb Filter [COMB]

A comb filter is a delay line with a feedback path around it. The delay length is tuned to the specified frequency you set. The filter response has peaks at the fundamental frequency (the one that directly corresponds to the delay line length) as well as at the multiples (harmonics) of the fundamental and notches between them. The feedback path gain controls the depth of the peaks and notches. You can adjust the *frequency* and *depth* of the comb filter, as well as select linear, cubic, or no comb filter delay tap position *interpolation*.

No interpolation means that the tap position snaps to the nearest sample position. Linear interpolation means that the tap position can fall between two consecutive samples and the output signal is calculated from the values of these samples as well as from position of the tap as related to the neighboring samples. This can offer much cleaner sound. Cubic interpolation introduces a higher order polyphase filter which uses five samples to calculate the output tap signal, offering higher quality than linear interpolation, but at the price of higher CPU load. When you will be basing high-quality modulation effects off the comb filter, we highly recommend you use *cubic* interpolation.

Phaser [PHASER]

Phasing is a classic modulation effect implemented as a cascade of low-order all-pass filters. The frequency response of the filter structure is flat, but its phase response is not.

When the resulting signal is mixed with the dry (input) signal the phase-shifts cause certain frequencies to be attenuated (cancelled) and the other to be amplified. The *feedback* parameter controls the phase response, which in turn determines the position of the notches on the frequency axis. The *depth* parameter adjusts the amount of dry signal to be mixed with the phase-shifted signal.

To make the phasing effect even more intense, create a feedback path to the Operator as we described in the section on the CFG matrix. The number of consecutive all-pass filters in chain can be adjusted between 1 and 16 by setting the *stage* parameter. The even number of filters creates the typical phasing effect, while the odd number of filters results in high-frequency damping as well, with the effect sounding like a combination of a phaser and low-pass filter.

Lo-Fi [LOFI]

If you are looking to capture that "trashy" and low fidelity sound, PSP Nitro includes a downsampler/bit-crusher Operator. Unlike most bit crushers and downsampler effects that are limited to selecting particular common bit values and sample rates, the PSP Nitro lo-fi plug-in gives you precise, continuous control over the *sample rate* and *bits* of your audio signal.

For sounds with rich harmonic contents, bit reduction sounds like adding the noise to the signal. If the audio signal is more like a single tone, the effect of bit reduction sounds like harmonic distortion rather than noise.

Reducing the sample rate omits every few samples, a process also known as decimation. Which samples are omitted depends on the relation between the original and resulting sample rate. If the original signal spectrum contains frequencies higher than half the resulting sample rate, aliasing will occur. This adds a distinctive anharmonic content to the signal's spectrum. Please note that even when the decimator output sample rate differs by small amount from your project's sample rate the aliasing will be clearly audible. The reason is that there is neither antialiasing filter nor interpolator in the lo-fi module.

Saturation [SAT]

The Saturation effect is a waveshaper with a symmetric transfer function—meaning that both positive and negative halves of the sampled wave are affected in the same way. Provided the input signal is a simple tone with no DC-offset, the odd harmonics will be added to it. If the wave's frequency content is richer than a simple tone (which is usually the case), inter-modulation occurs, resulting in more complex harmonics generation. This effect also causes the signal to be compressed, so the overall loudness can be increased. You can adjust the *shape* and *drive* of the Saturation effect.

Stereo Width/Balance [WID/BAL]

This Operator type allows you to adjust the stereo width of your audio expanding or reducing the stereo field of a stereo track. This Operator allows you to adjust the width from 0% to 200%, and the stereo balance continuously from hard left (L100) to hard right (R100).

L/R Independent Panning [PANNER]

Not only does PSP Nitro allow you to widen and adjust the balance of the stereo field, but this Operator type allows you to independently pan each channel of your stereo signal independently from the other. It offers two parameters, *Pan L* and *Pan R*, and each can be adjusted from hard left (L100) to hard right (R100).

Fractional Time Delay [GLIDE]

In the context of audio, the Glide Operator adds interpolated delays to the left and/or right audio signal. Technically, the Glide processor is a stereo delay line capable of generating delay times that correspond to a fractional number of samples (such as a 30.45 samples long delay, and so on). This allows for smooth modulation of the delay time resulting in high quality flanger, detune and pitch-twisting effects.

The Glide Operator lets you select the *mode* (milliseconds or tempo synced rhythm), the Left and Right *delay times* (from 0 to 4000ms in millisecond mode, or 8 bars to 1/64 triplet time in rhythm). Finally, you can choose no *interpolation*, linear *interpolation*, or cubic *interpolation*, (the effects of each type of interpolation are the same as the interpolation options in the Comb Filter (COMB) above).

LFO Section

PSP Nitro offers two low frequency oscillators (LFOs) that can be used to modulate many processing parameters of the plug-in. These are accessed from the LFO tab of the LCD Display. **Figure 18** shows the LFO section of the LCD Display.

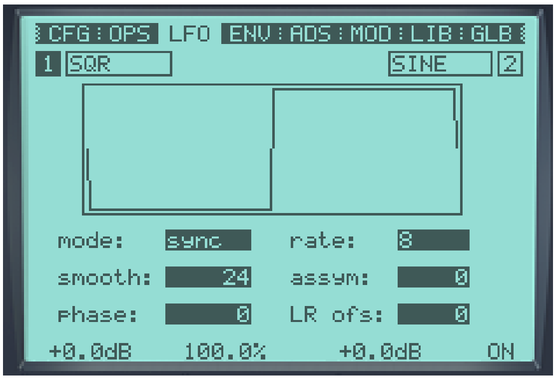


Fig. 18: The LFO screen of the LCD Display houses the LFO graph and parameters.

Clicking either number at the top of the LFO screen will select either LFO 1 or LFO 2 for editing. The text box next to each number displays the waveform of the selected LFO. Click in waveform text box to change the LFO waveform.

Types of Waveforms
Your waveform choices are:

- **SINE**: Sinusoidal waveform
- **TRI**: Triangular waveform
- **SQR**: Square waveform
- **SAW**: Sawtooth waveform
- **S&H**: Sample and Hold waveform

The parameters you can adjust for each LFO are explained below.

mode and rate
The *mode* you choose for each LFO determines the format of the *rate* control. The LFOs have 3 distinct modes for you to choose between:

- **freq:** In this mode, you specify the LFO rate in Hz.
- **rhythm:** This mode allows you to select a musically relevant rate (1/2 note, 1/4 note, and so on) based on the tempo of the host application.
- **sync:** This mode syncs the LFO phase to the start of the song. Unlike rhythm mode, in which the LFO is free-running, the LFO phase at a given song position will always be the same.
- **lfo1:** Only LFO2 offers this mode. While in this mode, LFO2's rate is expressed as a fraction of LFO1's rate. Both LFO's stay locked in sync.

smooth
Both LFO's offer a gentle low-pass filter to smooth out the discontinuities in the shape of the waveform. The cutoff frequency of this filter is controlled via the *smooth* parameter. Please

note that how the smoothing filter affects the actual waveform shape depends also on the LFO rate. The waveform shape plotted on the graph is always the actual shape.

assym

If you chose a square wave for the LFO's waveform, you will see this parameter (otherwise you will not). This parameter allows you to set the asymmetry, or the width of the waveform, between –100 and 100.

phase

When the LFO phase is synchronized to the song position, you might want to shift the phase of the LFO. The phase parameter allows you to adjust the phase between –180 degrees and 180 degrees.

LR ofs

The output of both channels of each stereo LFO is normally identical. You may adjust the LR offset parameter to create unique and creative stereo effects with left and right channels being modulated out of phase . If the Operator is in M/S (middle/side) mode, this will adjust the offset between the middle and side (instead of left and right) modulation signals.

ENV Section

The ENV screen of the LCD Display contains PSP Nitro's envelope follower parameters and graph. **Figure 20** shows the ENV screen.

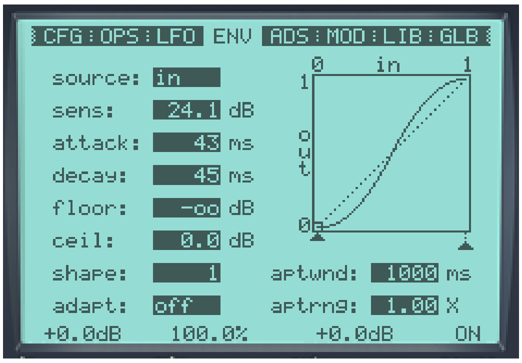


Fig. 20: The ENV section of the LCD Display contains the parameters and graph of PSP Nitro's envelope follower.

To use PSP Nitro's envelope follower, you need to attach it to a source. Then it analyzes the signal at that source and outputs the value that corresponds to the average signal loudness which can be used to modulate plug-in's processing parameter of your choice. The transfer function of the envelope follower shown on the graph has three adjustable parameters – floor, ceiling and shape and controls the relation of the signal loudness detected and envelope follower output .

The parameters of the envelope are explained below.

source
You have a choice of 6 possible sources: the input signal, output signal, and the signal at any of the four operators. If you wish the envelope to only respond to particular frequencies of your signal, set an Operator to filter just those frequencies, then set the envelope's source to that Operator.

sens
The sensitivity parameter allows you to preamplify or attenuate the signal before it is analyzed. This is useful for particularly soft or loud signals, accordingly.

attack/decay
The time constants used to calculate the average signal loudness is set via the attack and decay parameters. Both are defined as the time it takes the envelope follower to reach 70% of the step input level (which corresponds to –3dB).

floor/ceil/shape
The shape of the envelope curve is determined by the interaction of the floor, ceiling, and shape parameters. The *shape* parameter can be adjusted between 1 and 6; with each number representing an increasingly steep envelope curve shape. The *floor* parameter determines at

what dB the first envelope breakpoint will occur. The *ceiling* parameter determines at what dB the second envelope breakpoint will occur. The floor and ceiling parameters can be adjusted either by clicking the mouse in the box next to the parameter and moving the mouse up and down, or by moving the two triangles under the envelope graph.

adapt/aptwind/aptrng

You do not have to manually set the working range of the envelope (floor and ceiling levels). PSP Nitro can adaptively set them for you. To allow Nitro to configure the envelope floor and ceiling levels, turn the *adapt* parameter to on. PSP Nitro will then estimate the input signal average level and variation within certain time window, and use that information to set the floor and ceiling parameters. The length of time window PSP Nitro uses to estimate the input signal average level and its deviation is determined by the *aptwind* parameter. You can set this parameter to any number of milliseconds between 0ms and 10,000ms. The *aptrng* parameter then adjusts the width of the resulting range from .25x to 4x of the standard range.

ADS Section

The ADS section offers the parameters and graph for another of PSP Nitro's modulation sources: its ADSR envelope generator. **Figure 21** shows the ADS screen.

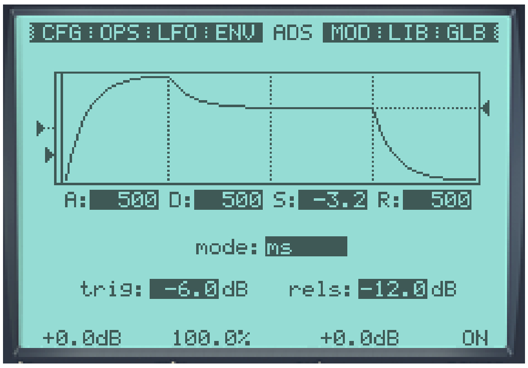


Fig. 21: The ADS screen contains PSP Nitro's ADSR envelope generator.

ADSR Segments
The ADSR generator is so named for its 4 segments:

- **attack** (the attack time of the envelope)
- **decay** (how quickly the envelope reaches its sustain level once the attack phase is over)
- **sustain** (level at which the envelope sustains)
- **release** (how quickly the envelope fades)

You can adjust these values by clicking in the text box to the right of the letters under the ADSR graph. The sustain parameter can be set either from the text box or the triangle to the right of the ADSR graph

mode
The format value for the ADSR can be switched between two *modes*:

- **ms** — the segments can be adjusted between 0 and 5,000ms
- **rhythm** — the time is host tempo-related, and the parameter can be adjusted in rhythmically relevant values.

In both modes, the sustain level can be set between -∞ and 0 dB..

trig/rels
You can select any modulation source in the MOD section to trigger the ADSR, but the ADSR will only be triggered when the signal is above the triggering level and released when it is below the release level of the ADSR generator. The *trig* and *rels* parameters control the triggering level and release level of the ADSR envelope, respectively. These parameters can be adjusted either by clicking in the box to the right of the parameter name and dragging the mouse up and down, or by adjusting the two triangles on the left side of the graph.

MOD Section

PSP Nitro's modulation system offers users nearly unlimited flexibility. You can choose from over a dozen modulation sources (over a hundred, if you count each MIDI CC as a separate source!), and route them to over fifty possible destinations! The modulation routing screen is accessed via the MOD tab of the LCD Display. **Figure 22** shows the basic modulation system screen.

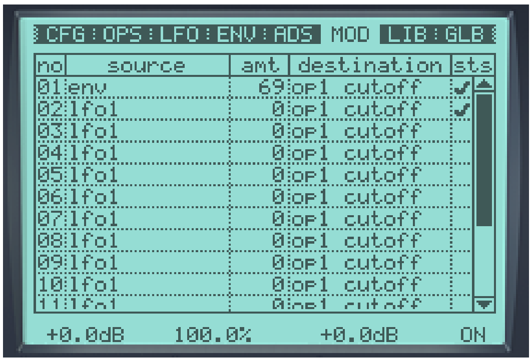


Fig. 22 The MOD tab on the LCD Display access PSP Nitro's powerful modulation section.

The basic concept of PSP Nitro's modulation system is very simple. First of all, you can connect up to 16 modulation sources to up to 16 different modulation destinations. The main modulation screen lays out, from left to right what you need to go about setting up a modulation connection.

Selecting A Modulation Source

For each modulation connection you need to make, first you need to select a modulation source. When you click in the *source* box next to one of the modulation connection numbers, the screen changes to the source selection screen, show in **Figure 23**.

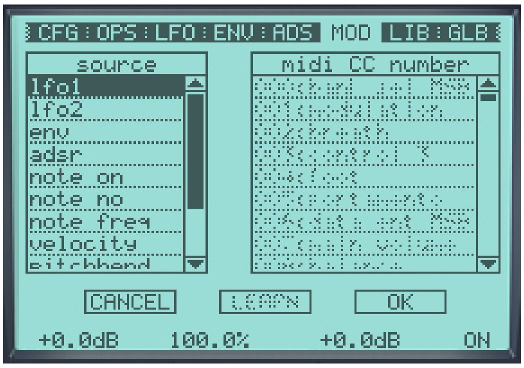


Fig. 23: When you click in a source box in the modulation screen, you will enter this source selection screen.

From here you can select any of the following 13 possible modulation sources:

- **lfo1**
- **lfo2**
- **env**
- **adsr**
- **note on** (maximum positive value when any note is on; 0 when no note is pressed)
- **note no** (value depending on the midi note number of the note that has been pressed)
- **note freq** (similar to note no, but the numbers are mapped to frequencies, so if the modulation amount is 100%, and the filter cutoff is set to minimum value (20Hz) the filter cutoff frequency will follow exactly the frequency of the note played
- **velocity**
- **pitchbend**
- **poly pressure**
- **aftertouch**
- **midi CC**

If you choose midi CC as your modulation source, the midi CC Number column at the right of the screen, and the learn button become available, as shown in **Figure 24**. In the midi CC Number column, you can select manually which midi CC number you wish to be the modulation source. If you are not sure of which midi CC number you wish to use, you can use the PSP Nitro LEARN button, which will detect a midi CC for you.

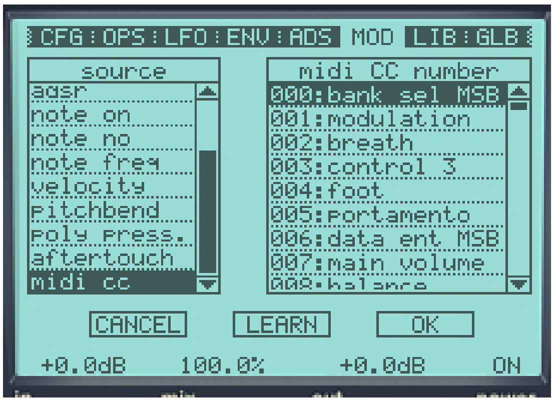


Fig. 24: If you select midi CC as your modulation source, you can then select a midi CC Number, or use the LEARN button for PSP Nitro to automatically detect which midi CC you are sending.

MIDI Learn

To use the LEARN button, click the button, and then move the MIDI Controller you wish PSP Nitro to respond to (for example, turn the knob or push the slider on your controller). Nitro will then record the MIDI CC number which the control you moved sends out.

After selecting your modulation source, click the OK button to return to the main modulation system screen.

Setting The Modulation Depth

The next step in making a modulation connection is to set the modulation depth. In the *amt* column, you can set the modulation depth from -100% to 100%.

Selecting The Modulation Destination

Once you have selected your modulation source and modulation depth, its time to select what it is that it will modulate. When you click in the *destination* column for that modulator, you will access the modulation destination screen shown in **Figure 25**. Here, all the parameters available for modulation are listed.

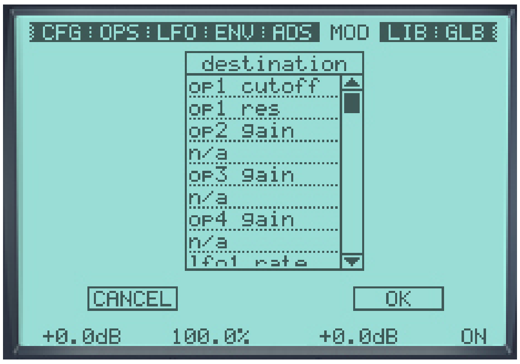


Fig. 25: In the modulation destination screen, you can choose from any of your available destinations.

The following modulation destinations will be available:

- The adjustable parameters from all Operators you have activated
- All LFO parameters for both LFOs.
- Input, main mix and output parameters,
- The values in all the boxes you have activated in the routing matrix
- The modulation depth of the active modulators
- The ADSR Trigger parameter

Modulation Status

The last step in activating the modulator is to change its status to on. The *sts* column allows you to quickly turn on and off each modulator. Click the box until there is a check inside; this indicates the modulator is on. If you click away the check mark, the modulator will be turned off.

NOTE: Most modulation sources are unipolar and positive with the exception of the LFOs which are bipolar. When you set up a parameter to be at a certain value (let's call it the 'original value') and then modulate it with a unipolar modulation source, the resulting value of the parameter will always be *equal to or greater than* the original value (or *equal or less than* the original value if the modulation depth is negative). In contrast, when the original parameter is modulated by a bipolar modulation source (such as LFO 1) its resulting value will *oscillate around* the original value.

LIB Section

PSP Nitro uses its own proprietary preset management system. This allows for interchange of presets between different platforms and plug-in formats regardless of host application. This also makes it very easy to move presets between projects. The PSP Nitro preset management system is accessed via the LIB tab on the LCD Display, as shown in **Figure 26**.

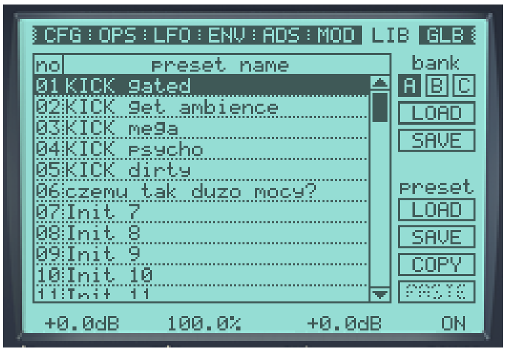


Fig. 26: Selecting the LIB tab of the LCD Display allows you to save, load, and copy presets between PSP Nitro's three preset banks.
We strongly recommend using the built-in preset system instead of the one provided by host. Most host application will only see one preset and it will not allow to perform the operations other than just changing the preset's name.

Bank Section
PSP Nitro offers 3 banks that can contain up to 64 presets each. You can load any bank into any of the three available banks by selecting that bank and pressing the LOAD button. You will be presented with a file dialog window in which you can choose the to bank to load. After choosing a bank file, the selected bank will be loaded into the Bank (A, B, or C) that you have selected. You can activate any preset in the loaded bank by selecting it in the preset name column. You can rename any selected preset by clicking again on its name.

You can also choose to SAVE the loaded bank to disk by pressing the SAVE button, at which point you will be prompted with a file dialog window to choose a name and destination for the bank you wish to save. We recommend that you store all your Bank files in the same location. This makes it easier to find them later.

Preset Section
You are not limited to only loading and saving entire banks. You can use the LOAD and SAVE buttons in the Preset section to only load or save single presets. The loaded preset will replace any preset in the selected preset slot in the currently loaded bank. If you wish to copy a preset from one bank to another, press the COPY preset to copy the preset into memory, load the new bank, then PASTE the preset into a slot in the newly loaded bank.

Factory Presets
Nitro comes with 192 factory presets. Some of them have been designed for processing instruments and vocals, some are for loops processing purposes.

It is worth noting that many presets use song position synchronization for the LFOs. If the song is not being played or if the host is not capable of sending the song position information to the plug-in the LFO will not run and the preset will not process the sound the way it was designed to. Always look at the LFO1/2 mode to see if the preset uses the song position synchronization.
Most of the presets using the envelope follower set the adaptive mode to on to sound perceptible on most audio. As it takes the envelope follower some time to settle while in adaptive mode, we always suggest setting the floor and ceiling level manually to match the audio being processed.
Last but not least, there are few sidechained presets. They all assume the sidechain signal to be fed into the left input channel and the main signal to the right.

GLB Section

The final section of the LCD Display contains global options that affect all of PSP Nitro's operations. These parameters are accessed by selecting the GLB tab of the LCD Display, as shown in **Figure 27**. The GLB section also displays which version of PSP Nitro you are using, as well as your license. Always check the PSP Audioware Website for the latest updates to your PSP Nitro plug-in.

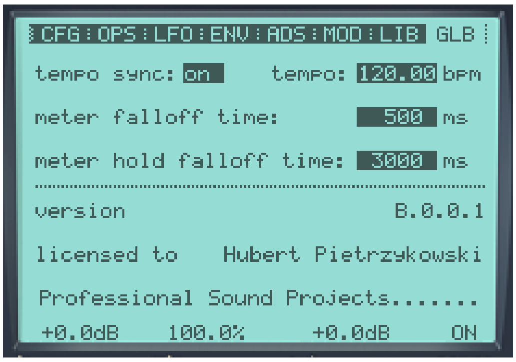


Fig. 27: The GLB Section of the LCD Display contains options that effect PSP Nitro globably.

A short description of each parameter follows.

Tempo Sync
If tempo sync is set to on, the PSP Nitro tempo will be automatically set to match your project tempo, provided the host is capable of sending the tempo information to the plug-in.. If tempo sync is set to off, the tempo is set manually.

Tempo
If your host does not offer tempo sync or it is turned off, you may set the tempo manually here. If the tempo sync is on the tempo will reflect the tempo of your project and automatically follow its changes.
The tempo affects those plug-in parameters that offer synchronization with the host tempo which are:

- GLIDE operator time when in rhythm mode
- LFO rate when in rhythm or sync mode
- ADSR attack, decay and release time when in rhythm mode

Meter Falloff Time
This sets the time it takes the plug-in level meters to refresh after showing a given maximum value. This number can be adjusted between 1ms and 1000ms.

Meter Hold Falloff Time
The PSP Nitro Meters also show the peak levels. Just as the previous parameter this one determines how fast the indicator will fall from maximum value. It can be adjusted between 500ms to 5000ms, or HOLD, which will keep the last peak value until the user clicks on the level meters.

Support

To receive free technical support, news about new products, updates, upgrades and special offers please the register your PSP product our web page:
http://www.pspaudioware.com/registration/register.htm

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